

REMARKS

By this Amendment, claims 1, 5 and 7 have been amended. Claims 1-9 remain pending in the present application.

Claims 5 and 7 have been objected on the ground that the claims are duplicates of claims 4 and 6. To address the Examiner's concern, the dependencies of claims 5 and 7 have been amended to each depend from claim 2, as was originally the case. Since claims 4 and 6 each depend from claim 1, this objection has been overcome, whereupon withdrawal of the same is respectfully requested.

Claim 1 has been rejected under 35 U.S.C. § 102(e) as being anticipated by Yashima et al., U.S. Patent No. 5,953,431.

The present invention as recited in claim 1 includes "a processor for comparing in real time an output signal from the microphone with an output signal from a sound source with reference to a frequency characteristic and an echo characteristic of the sound regenerated from the loudspeaker, or a reverberation characteristic of the sound, including the delay time for the echo characteristic or the reverberation characteristic, and correcting a signal from the sound source using the difference in output signal between the microphone and the sound source by reference to the frequency characteristic and the echo characteristic or the reverberation characteristic."

Yashima falls far short in meeting Applicant's claimed invention, as Yashima discloses a device which determines coefficient data based on a comparison between a noise signal from a noise source and an output from a microphone, to correct a sound pressure frequency at a listening position. Yashima fails to compare a microphone output signal and a sound source output signal with reference to a frequency characteristic and an echo characteristic of the sound regenerated from the speaker, or a reverberation characteristic of the sound, including the delay time thereof, and correcting the sound source output signal based on the difference with reference to the same characteristics.

The Examiner contends that "echo and/or reverberation characteristics are inherent [in Yashima's disclosed frequency characteristics]." The Examiner's assumption, however, is simply

not true. The sound pressure frequency discussed in Yashima relates to the number of sound pulses per unit of time. An echo is defined as the reflection of sound off a surface which thereby produces a secondary sound effect. Similarly, a reverberation is defined as a series of echoes, or re-echoes of the original sound. Clearly, then, consideration of echo and/or reverberation characteristics require consideration of not only the frequency of the sound, but also information such as the size of the room, the absorption of sound within the room, etc., for example. Such characteristics can never be "inherent" with the frequency characteristics of sound. Thus, absent any specific mention in Yashima's disclosure, there is no teaching in this patent of the claimed comparison "with reference to a frequency characteristic and an echo characteristic of the sound regenerated from the speaker, or a reverberation characteristic of the sound."

Furthermore, unlike the claimed invention, Yashima is completely silent as to the consideration of any delay times with respect to any echo or reverberation characteristics. The delay circuit 56 disclosed in Yashima serves merely to compensate for the delay time of the signal traveling through the system components (col. 13, lns. 9-15), and is not related to any delay time which may be associated with any echo or reverberation characteristics of the sound outputted through speaker 60.

Further still, in view of its failure to calculate the difference signal with reference to the echo or reverberation characteristics, Yashima's system is merely akin to the prior art discussed in the Background section of Applicant's specification, in which the correction process occurs by switching a switch to a reference source (in Yashima's case, the reference source is a noise source), and comparing a difference between the microphone output signal and the reference source (Yashima, col. 12, lns. 46-54; col. 13, lns. 16-28).

In view Yashima's failure to teach each and every feature of Applicant's invention as recited in claim 1 as highlighted above, Yashima cannot anticipate the claimed invention. Accordingly, withdrawal of this rejection is respectfully requested.

Claim 1 has been rejected under 35 U.S.C. § 102(b) as being anticipated by Rao et al., U.S. Patent No. 6,141,415.

As a preliminary matter, Applicant notes that the issue date of the Rao patent is October 31, 2000, which is after the U.S. filing date of the present application. Accordingly, Rao does not

As a preliminary matter, Applicant notes that the issue date of the Rao patent is October 31, 2000, which is after the U.S. filing date of the present application. Accordingly, Rao does not qualify as a § 102 (b) reference against the present application.

Rao teaches a speaker-phone type system in which each of two ends serves as a transmitter and receiver of sound, to thereby enable two speakers, one at each respective end of the system, to hear the sounds uttered by the other speaker, without also hearing his or her own speech mixed in with the sounds from the other end upon being transmitted through the loudspeaker at the other end. The problem addressed by Rao's system is described at column 1, lines 25-38.

To best illustrate how the claimed invention distinguishes from the system disclosed in Rao, assume that the far end speaker 42 in Fig. 2 in Rao corresponds to the sound source for the present invention, and Rao's speaker 75 corresponds to Applicant's claimed loudspeaker, and Rao's microphone 80 corresponds to Applicant's claimed microphone. Based on this perspective of Rao's system, sound is produced by the far-end speaker 42 and is translated into sound 76 outputted from the speaker 75 in Rao (see, *e.g.*, col. 3, lns. 20-27). Microphone 80 picks up the sound 77 produced from speaker 75, and also sound produced by a near end speaker 45. Thus, the output signal from the microphone represents a mixture of sounds originating from both ends 42, 45 of the system.

The output signal from microphone 80 is sent to adder 64 via preamplifier 78. Meanwhile audio echo cancellation circuit 62 "receives a portion or sample of the output of" the circuit 55, which is also outputted to speaker 75 as the sound originating from far-end speaker 42. In other words, audio echo cancellation circuit 62 captures the sound produced by the far-end speaker 42, and sends this sample to the adder 64 (col. 6, lns. 45-47). Adder 64 then subtracts the signal provided by audio echo cancellation circuit 62 from the signal from the preamplifier 78. In other words, the output signal from the microphone is corrected by subtracting the signal from AEC 62 (col. 6, lns. 47-48).

Applicant's claimed invention differs vastly from Rao in that the signal from the sound source, and not the output signal from the microphone (as is the case in Rao) is corrected based on the difference data obtained from the comparison, as recited in claim 1 in the present

application. Even if the operation perspective of the system disclosed in Rao were switched, so that near end speaker 45 is considered to be the sound source and the speaker and microphone is placed at the end of far end speaker 42, the result is the same, using adder 63 and LEC 63, instead of adder 64 and AEC 62, respectively.

Since Rao does not teach each and every feature of the invention as recited in claim 1, the claimed invention cannot be anticipated by Rao. Accordingly, withdrawal of this rejection is respectfully requested.

Claim 2 has been rejected under 35 U.S.C. § 103(a) as being unpatentable over Yashima in view of Kuusama et al., U.S. Patent No. 5,559,891.

As demonstrated above, Yashima is insufficient to meet the claimed invention recited in claim 1, from which claim 2 depends. The features of the claimed invention found lacking in Yashima are also absent in Kuusama, as will be apparent from the discussion below. Hence, the claimed invention cannot be rendered obvious by a combination of Yashima and Kuusama.

Kuusama teaches a sound system including an ambience generator which purposely adds delayed signals to the sound emitted from the speaker to produce a desired resonance. Nowhere in the reference does Kuusama disclose comparing in real time an output signal of the microphone with an output signal from a sound source, much less with reference to a frequency characteristic and an echo characteristic of the sound regenerated from the speaker, or a reverberation characteristic of the sound, including the delay time thereof, and correcting the sound source output signal based on the difference with reference to the same characteristics, as claimed in the present application.

In addition to the features of Applicant's claim 1 not found in Kuusama (or Yashima), Kuusama (and Yashima) also fails to disclose a first A/D converter for converting a signal outputted from a sound source, and also a second A/D converter for converting a feedback signal outputted from a microphone, a memory for storing digitized voice sample data taken within a fixed time determined as a subject time for the delay of the reverberation and echo of the sound, or a successive comparison analyzer for comparing the feedback data with the stored data to analyze the intensity of reverberation and echo, as recited in claim 2. Although Kuusama discloses a data memory 22, there is absolutely no teaching in the reference that the memory 22

stores digitized sound data taken within a fixed time determined as the subject time for the delay of the reverberation and the echo, as recited in Applicant's claims.

In view of the foregoing, withdrawal of the rejection under § 103 is respectfully requested.

Dependent claims 3-9 each depend ultimately from claim 1, and therefore incorporate each of the features recited therein. As such, claims 3-9 are also patentably distinguishable over the cited prior art for at least the reasons discussed above with respect to claim 1.

As all of claims 1-9 have been demonstrated to be allowable over the prior art of record, Applicant further submits that the present application is currently in condition for allowance, whereupon early and favorable reconsideration in this regard is courteously solicited.

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Respectfully submitted,

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APPENDIX B
VERSION WITH MARKINGS TO SHOW CHANGES MADE
37 C.F.R. § 1.121(b)(iii) AND (c)(ii)

SPECIFICATION:

Replacement for the paragraph beginning at page 2, line 4:

Further, with the loudspeaker unit [of] in which only [the] correction the frequency characteristic is executed, there is a problem in that no correction can be made to a sound lag and phase shift [to be] caused by the reverbation and [an echo] echoing of [a] the sound.

Replacement for the paragraph beginning at page 2, line 21 through page 3, line 8:

The loudspeaker unit of the present invention [adapted to the environment] comprises a microphone for picking up a sound regenerated from a loudspeaker; processing means for comparing [at] in real time an output signal from the microphone with an output signal from a sound source [with reference to] , in particular by referencing the characteristic at an optional frequency and the [characteristic] characteristics of the reverberation as well as of the echo, each including [the] a delay time, respectively, and correcting [a] the signal from the sound source with the difference output signal between the microphone and the sound source; an amplifier for amplifying the output of the processing means; and a loudspeaker unit.

Replacement for the paragraph beginning at page 3, line 9:

Also in the present invention, [it is allowable to correct] a signal to be sent to the loudspeaker is corrected by the result [learned] obtained through an arithmetic operation. [It is acceptable to intermittently renew the] Using the result of a comparison operation, a parameter which is used to correct the signal to be sent to the loudspeaker [by using the result of the comparison] is intermittently renewed.

Replacement for the paragraph beginning at page 4, line 2 (through line 8):

In other words, according to the present invention, the loudspeaker unit [can save] does not need a reference signal generator to be used for comparison and a switch for selecting this signal, unlike the prior art.

Further, since the processing module of the loudspeaker unit [catches] receives a feedback signal [at] in real time, the [particular] procedure described above in the prior art loudspeaker unit is not needed for the correction.

Replacement for the three paragraphs beginning at page 4, line 14:

Fig. 2 is a structural view showing an embodiment of a loudspeaker unit adapted to the environment [of] in accordance with the present invention.

Fig. 3 is a structural view showing a concrete embodiment of a loudspeaker unit adapted to the environment [of] in accordance with the present invention.

Fig. 4 is a structural view showing another embodiment of a loudspeaker unit adapted to the environment [of] in accordance with the present invention.

Replacement for paragraph beginning at page 8, line 3:

The intensity of the reverberation and the change of the frequency characteristic are corrected according to the result [learned about] obtained with respect to the data of sound source 2. After [clearly grasping] analyzing the frequency characteristic and the delay [of] attributable to the reverberation as well as the echo, the value set for correction is changed to determine the correction parameter.

Replacement for the two paragraphs beginning at page 8, line 12:

With reference to Fig. 4, in order to decrease the load of processor module 3, data processing for the correction [purpose] process is not [to be] performed [at] in real time, but a correction parameter previously extracted from the past [example] feed back iteration is [better] used intermittently[, and thus]. Thus, it becomes possible to correct the sound delay and the

phase shift which may be caused by [the] reverberation and [the] echo of the sound.

Further, by attaching microphone 6 to a casing of loudspeaker unit 1 of the present invention, [the] any wiring [to be laid outwardly from] which would otherwise be exposed outside of the casing can be omitted.

CLAIMS:

1. (Amended) A loudspeaker unit for a sound source, the loudspeaker unit being adaptable to changing environments, comprising;

a loudspeaker;

a microphone for picking up sound regenerated from [a] the loudspeaker;

5 a processor for comparing in real time an output signal from the microphone with an output signal from a sound source with reference to a frequency characteristic and an echo characteristic of the sound regenerated from the loudspeaker, or a reverberation characteristic of the sound, including the delay time for the echo characteristic or the reverberation characteristic, and correcting a signal from the sound source using the difference in output signal between the microphone and the sound source by reference to the frequency characteristic and the echo characteristic or the reverberation characteristic; and

10 an amplifier for amplifying the output of the processor[; and a loudspeaker].

5. (Amended) A loudspeaker unit adapted to the environment according to Claim [1] 2, wherein, the frequency comparison of the characteristic and the comparison of the characteristic of the echo or the reverberation each including the delay time are learned by arithmetic and a signal to be sent to the loudspeaker is corrected according to the learned result.

7. (Amended) A loudspeaker unit adapted to the environment according to Claim [1] 2, wherein, the frequency comparison of the characteristic and the comparison of the characteristic of the echo or the reverberation each including the delay time are intermittently performed and a signal to be sent to the loudspeaker is corrected according to the comparison result.